Toward Assessing Subjective Quality of Service of Conversational Mobile Multimedia Applications Delivered Over the Internet: A Methodology Study

Pascal Dugénie, Alistair T. Munro, and Michael H. Barton

***Abstract*—Some recent publications have proposed methodologies to assess the performance of multimedia services in introducing subjective estimate of the end-to-end quality of various applications. As a general statement, in order to obtain meaningful subjective results, the experiments must be repeatable and the elements of the whole chain of transmission between users must be restricted to a minimum number of objective quality metrics. This paper presents the approach to specifying the minimum quality required by the deaf when using a sign language videotelephony application over the Internet with standard off the shelf equipment.**

***Index Terms*—Deaf, mobile, multimedia, sign language, video.**

I. Introduction

THE introduction of distributed multimedia applications opens new opportunities for human communication in the Global Information Society, in areas such as education, work, healthcare, entertainment, information, and leisure. Realizing these opportunities is assisted by the trend toward convergence of the elements of technology and communications supporting these services, coupled with increasing liberalization and automation that allow new services to be set up rapidly on demand.

The social opportunities are as important as the technical ones. It has been recognized widely for many years that a significant minority of the population is poorly served by existing telecommunications services, to the extent of being excluded entirely because the services are effectively unusable. A converged technology and communications platform can make applications accessible anywhere, anytime. The challenge of making them inclusive: accessible to and usable by *anybody*, is still to be addressed. Specifically, the ability to reconfigure the application components and modify the quality of service (QoS) constraints on underlying services to suit individual preferences and abilities is a key enabler for solutions to the *anybody* dimension. A case study presented in this paper provides a first motivating example: deaf people using H.261 videotelephony for sign-language conversation rely on accurate timely transfer of the upper-body motion associated with signing. High-resolution and high-refresh rate of the video streams will help them as much as it helps other users but the higher level of motion will tend to generate a greater level of traffic.

The same application will behave in different ways in terms of transmitted traffic for different people and place thereby different loads on the supporting communications infrastructure. This does not present a problem for conventional fixed network circuit-switched telephony using pre-allocated constant bit-rate synchronous fixed-delay channels (e.g., narrow-band ISDN). The ability of the network to transport the service will be compromised if any of these operational assumptions are changed. Two factors that make these changes inevitable are the migration toward the Internet as a universal packet-mode transport network and the integration of terminal mobility (where the network connectivity varies on a short time scale). The consequences of these changes form a second motivation for the work presented here: it will become much more critical to place bounds on the variation of traffic associated with personalized services under this new regime. We use effective bandwidth (EB) as the primary metric for the traffic bounds.

The final motivation is the presumption that users will be able to modify QoS constraints on demand. This implies that they will be able to make judgements about quality and adjust the configuration themselves.

This paper discusses an approach to drawing these threads together. The main objective is to define a methodology for estimating the EB consumed by a conversational multimedia application communicating in packet mode, relating the use of network resources to the subjective evaluation of the quality of the delivered service. The methodology is presented in Section III and the case study is described in Section IV. The analysis and evaluation are described in the following three sections with conclusions in Section VIII.

II. Background

A. Evolution of Communications Systems

Several technological trends must be considered.

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- No single underlying network transmission technology will be used, possibly changing during a session[1] if users are mobile or if user demands and personal environment change.
- Convergence of the domains of IT, telecommunications, broadcast, and consumer electronics will introduce a multiplicity of new services transported using these common transmission infrastructures. The rate of change of services is likely to be greatly accelerated as methods for rapid installation mature.
- Terminals capable of acquisition and display of a wide variety of data types must be able to adapt to a wide range of user needs and ambient conditions. Following the trends of convergence noted above, it is unlikely that such terminals will be designed in the traditional telecommunications paradigm but will, instead, be composed on demand from any suitable components accessible in a local personal communications space.
- Full-duplex interactive conversations introduce dependencies that will have an effect on traffic distribution during a session.

It is generally accepted that the Internet and its protocols (referred to henceforth as IP) will be the unifier of these dimensions of technological diversity. In fact the migration to IP for control and management is gaining momentum rapidly and will offer long-term rationalization. More significantly, it will support the integrated approach whereby in-session requests for changes to the parameters associated with communications links (termed QoS requests) can be propagated between the terminations of the session and to the underlying service provider. Advanced reactive and dynamic optimization algorithms and protocols can then be executed to acquire and modify the underlying resources to meet such QoS demands.

Such methods may succeed or they may fail. For example, the widespread uptake of wireless mobile telephony has led already to an acknowledgment that sessions may be terminated due to handoff failures or lack of coverage (see Section II-C). Acceptable quality during the session will be variable, e.g., distortions to voice quality, discontinuous operation, or variable network delay affecting echo-cancellation. We must now accept in addition that users may opt independently to change the characteristics of their session in ways that cannot be accommodated by one or more of the other collections of resources participating in that session.

In parallel, there is a move to replace synchronous end-to-end circuit-mode transmission with connectionless packet mode communication. Clearly this is essential for the IP everywhere approach. It is arguable that all significant advances in multimedia applications will be made using the Internet in any case. For wireless mobile communications in particular, where resources are limited, some of the ambitions for UMTS cannot easily be met if resources are tied up continuously to support conversations that exhibit a significant degree of burstiness. Thus, a packet mode approach that took account of burstiness could offer an opportunity to exploit any underlying statistical multiplexing gain.

[1]We use session to mean a relationship between two or more parties using such applications for real-time full-duplex communication.

B. Subjective Aspects of Quality of Service Assessment

Assessing the extent to which QoS demands are met satisfactorily requires a framework for measuring the subjective aspects of the service. The mean opinion score (MOS) is well established as a measure of telephony service quality for voice. However, this is only one dimension of quality and applicable in one well-defined context. When, for example, the application is interactive videoconferencing new MOS metrics will have to be devised. This has been researched extensively by Wolf and others [1]–[3], who have proposed methods that link objective measurements with subjective perceptions for MPEG-1 and MPEG-2 streams delivered over standard synchronous channels. ISO [20] and the ITU [17] have also developed generic standard methods.

In establishing these new MOS metrics and quantifying the subjective aspects, it will be important to consider *who* is being asked. There are emerging social issues that will influence this. In Europe and the United States, for example, universal access to telecommunications services will become a requirement to be implemented by operators and service providers under regulatory control. Therefore, the one in ten (or more) people who cannot use these services will have to be consulted [4]. This group includes those with impairments to mobility, motor and sensory capabilities. We will call these the *disabled*, following the international convention, while recognising that disability is difficult to define. People may have permanent impairments of sight, hearing, mobility, or cognitive capabilities, or they may be temporarily disabled—a noisy railway station, or someone trapped by a fire in a smoke-filled room. They may have multiple disabilities, which may increase with age—or, indeed, decrease for a growing child or somebody undergoing rehabilitation. The issue of accessibility and acceptability for the disabled has therefore many manifestations. One of these is the operation of the communications application under unconventional circumstances, which may generate system configurations that differ quite significantly from those hitherto encountered to achieve typical MOS values.

C. Linking Perceived Quality of Service With Network Grade of Service

It is possible to devise and measure aspects of *subjective perceived quality* independently of the causes of this perception. For example, loss of encoded voice samples during a digital cellular telephone call can be perceived as clicks or a warbling effect on the speaker's voice up to a certain level of loss; beyond this level there will be gaps in the speech flow. The criteria taken into account to evaluate a service will vary with the nature of the application. For example, loss of video samples will have a different perceived effect from loss of audio.

Such loss may occur at several different points on the path from the source network access point through to the playout device:

- in the transmission path up to and including the receiving network access point through corruption or other reasons for discard (e.g., lack of buffer capacity);
- at the receiving codec if the jitter in delivery places the sample outside the acceptable delay window;
- in the path between codec and playout device due to other system activity.

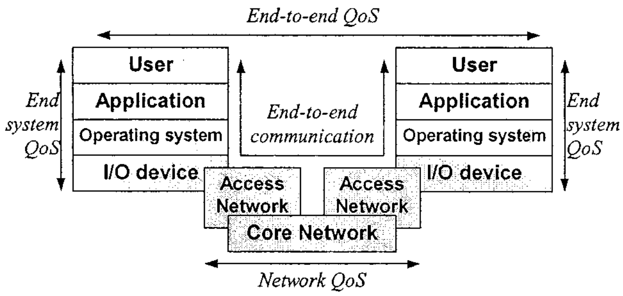

Fig. 1. Elements influencing end-to-end QoS.

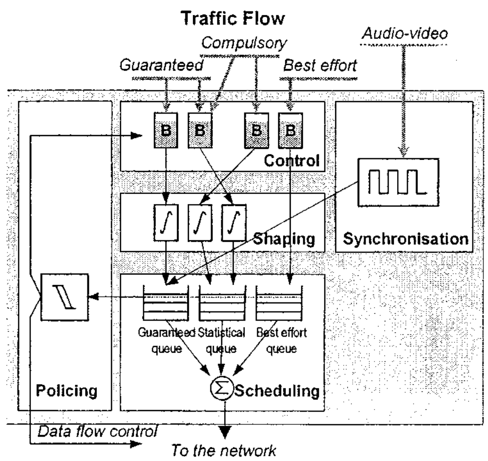

Fig. 2. QoS control mechanisms.

Such transmission considerations are taken into account in the engineering of a service provided by a typical telecommunications network and the type approval procedures for terminal equipment and other elements involved in the application. However, the influence of personal mobility, use of general purpose computing devices as terminals, wireless transmission media, and packet mode delivery using best effort IP, have focused attention on the increasingly unreliable base (at least in traditional terms) on which mobile multimedia applications will be offered. According to a review of previous work in this area [5], two approaches have emerged:

- *End-to-End QoS*—in which all key elements of the chain between users play a role, as shown in Fig. 1. Aurrecoechea [14] and Hutchison [15] have surveyed the well established architectures and compared the different approaches. They identify control mechanisms as illustrated in Fig. 2.
- *Evolution of Traditional Traffic Engineering Models*—the ETSI Technical Report ETR003 [8] reviews the factors influencing the perceived QoS such as the *communication establishment delay*, the *probability of blocking*, and the *EB*.

The two first factors are specified for circuit-switched services in the ITU rec. E.771 [13] and more generally in the E.750 series [11], [12]. The communication establishment delay is defined as the time interval from the instant the user initiates a connection request until the complete message indicating call disposition is received at the calling terminal. The probability of end-to-end blocking provides a global estimation of the link performance. Unsuccessful call attempts can occur at the radio link, at the interworking units or at the transit network because of a lack of resources either at the user plane or at the control plane.

The concept of EB has been developed over recent years to provide a measure of resource usage, which adequately represents the tradeoff between sources of different type, taking account of their varying statistical characteristics and the QoS requirements. This parameter has been analyzed in detail in the following section.

D. Effective Bandwidth (EB)

1) Definition: Previous work by Kelly [9] and Gibbens [10] has explored the use of a surface, $\alpha(s,t)$, as a representation of EB, where s is the *space scale* (typically in bits^{-1}) and t is the *time scale* (typically in seconds). Gibbens has evaluated α for two real broad-band traffic data sets: the *Star Wars* movie data set and the Bellcore Ethernet data set and has attempted to find analytical models to fit the results. Both the traffic source and the characteristics of the channel determine the appropriate time scale t and space scale s.

For the purpose of the present study, we investigated the form of the EB surface, estimated from data obtained from an application of sign-language videotelephony. Using the notation in [10], in a trace of N real traffic measurements, each packet is assumed to have a corresponding record giving the packet's time of arrival t_i and size x_i. It can be represented by the collection

$$X\{(t_i, x_i)\,;\; i = 1, \ldots N\}.$$

Considering the amount of data X arriving during the interval $I[\tau, \tau + t]$ to be

$$X[\tau, \tau + t] = \sum_{i=1}^{N} x_i I\left(\tau \leq t_i \leq \tau + t\right), \quad 0 \leq \tau \leq t_N - t$$

the EB is given by

$$\alpha(s,t) = \frac{1}{st} \log \frac{1}{t_N - t} \int_0^{t_N - t} e^{sX[\tau, \tau+t]} d\tau.$$

2) Properties: This definition begets four properties that are described in detail in Kelly's work [9]. The *property (iv)*, which is of particular interest in the present study, states the fact that *for any fixed value of* t*,* $\alpha(s,t)$ *is increasing in* s *and lies between the mean and peak of the arrival rate measured over an interval of length* t*. The form of* α *near* $s = 0$ *is determined by the mean, variance and higher moments of* $X[0,t]$*, while the form of* α *for high* s *is primarily influenced by the distribution of* X *near its maximum*. Fig. 3 illustrates this property.

In a practical context, t corresponds to the shaping mechanism factor and can be set either to a high value to limit the EB or to a low value to decrease the packet transfer delays. For sources containing periodicity, Kelly showed that there are pertinent values of t, for which a variation of s has a minimum effect on α. Evidence of this will be shown in the analysis of the traffic sources of our case study.

III. Methodology

Before beginning the evaluation of an application in a networked environment, one has to determine the origins and the

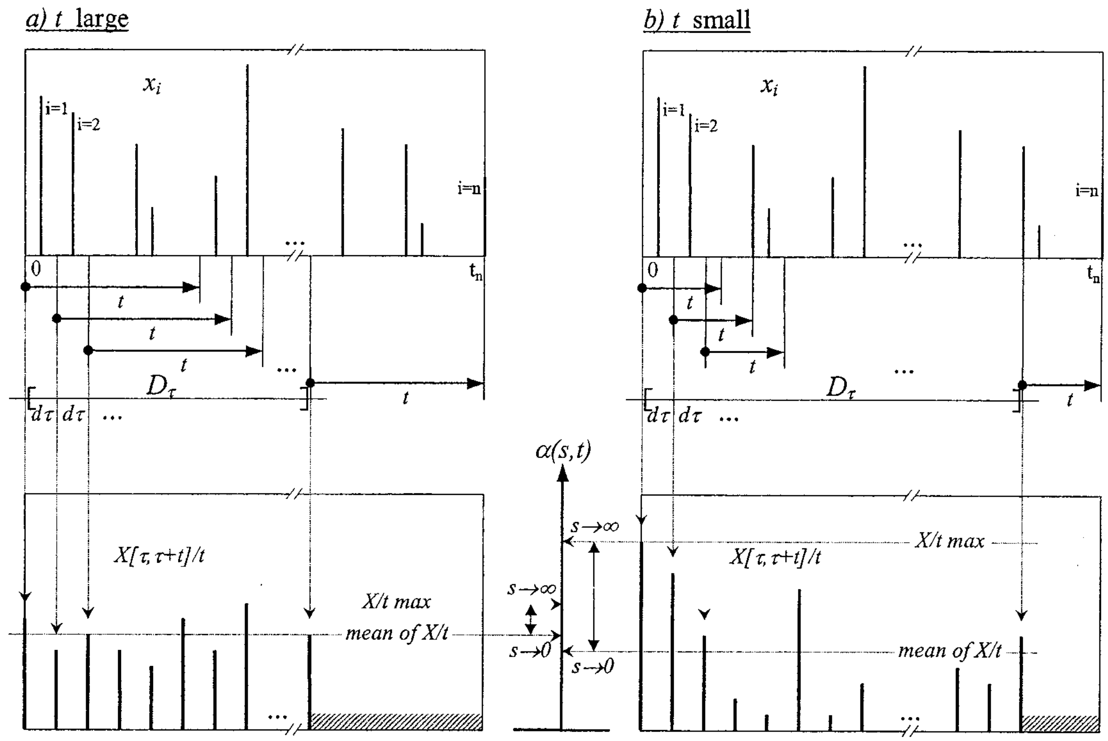

Fig. 3. Illustration of the influence of t and s on α. For larger t, α is distributed in a small interval for extreme variation of s, but this interval increases as t decreases.

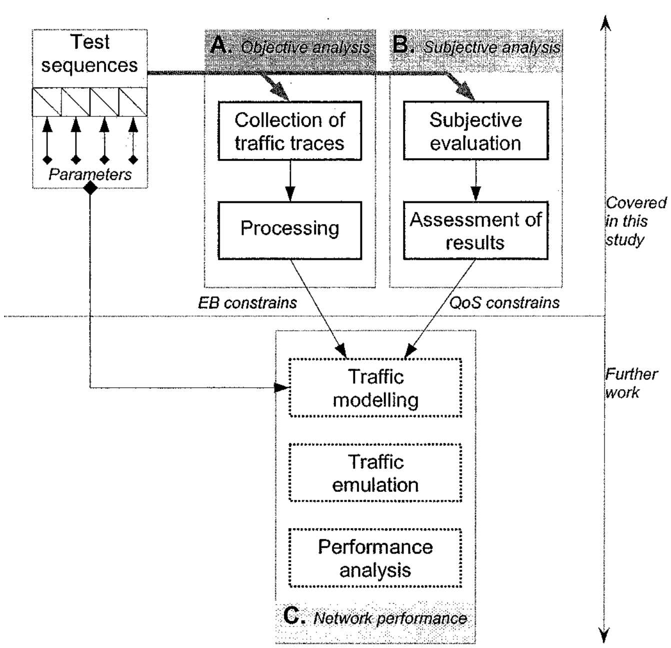

Fig. 4. Methodology to estimate the EB of an application.

parameters that will have a dominant effect on the end results. For instance, perceived slow motion in a videotelephony service is not only originated by a congested transport network; a slow video codec would cause a similar effect.

The methodology here aims to determine the EB required for the lower and the upper bounds of an application such as our case study described in Section IV. The procedure for measurement is inspired by the ISO recommendations for evaluation and testing of MPEG-4 codecs [16]. As shown in Fig. 4, it combines collections of objective and subjective data. The objective analysis is intended to determine the relation between relevant application parameters and the EB whereas the subjective analysis is intended to correlate the variation of these parameters with the user's perception of QoS.

Further quantitative cross-analysis will allow us to generalize the effects of the initial source parameters on the traffic shape, taking into account the EB and QoS constraints.

IV. Case Study: Sign-Language Videotelephony

A. Why This Application?

Videotelephony is a valuable application that brings many potential benefits to users who rely on visual means of communication.

The modalities of mute conversations are significantly different from the conventional vocal ones [6]. Gestures and facial expressions involving the whole upper body form part of the conversation. Signing and lip reading have their own human protocols that substitute for the absent audible cues. Motions vary from a single finger flex to an arm rotation movement and are continuous while the participant is communicating. A complete message can range from one single gesture or facial expression to a more complex combination of both of them. There is undoubtedly a large level of redundancy in the information flow, although this has not been measured as far as we are aware.

B. Quality Assumptions to be Tested

Because one or more different parts of the upper body may be active at any one time, the refresh rate of the scene must be high enough to convey the detailed motions. Synchronization of

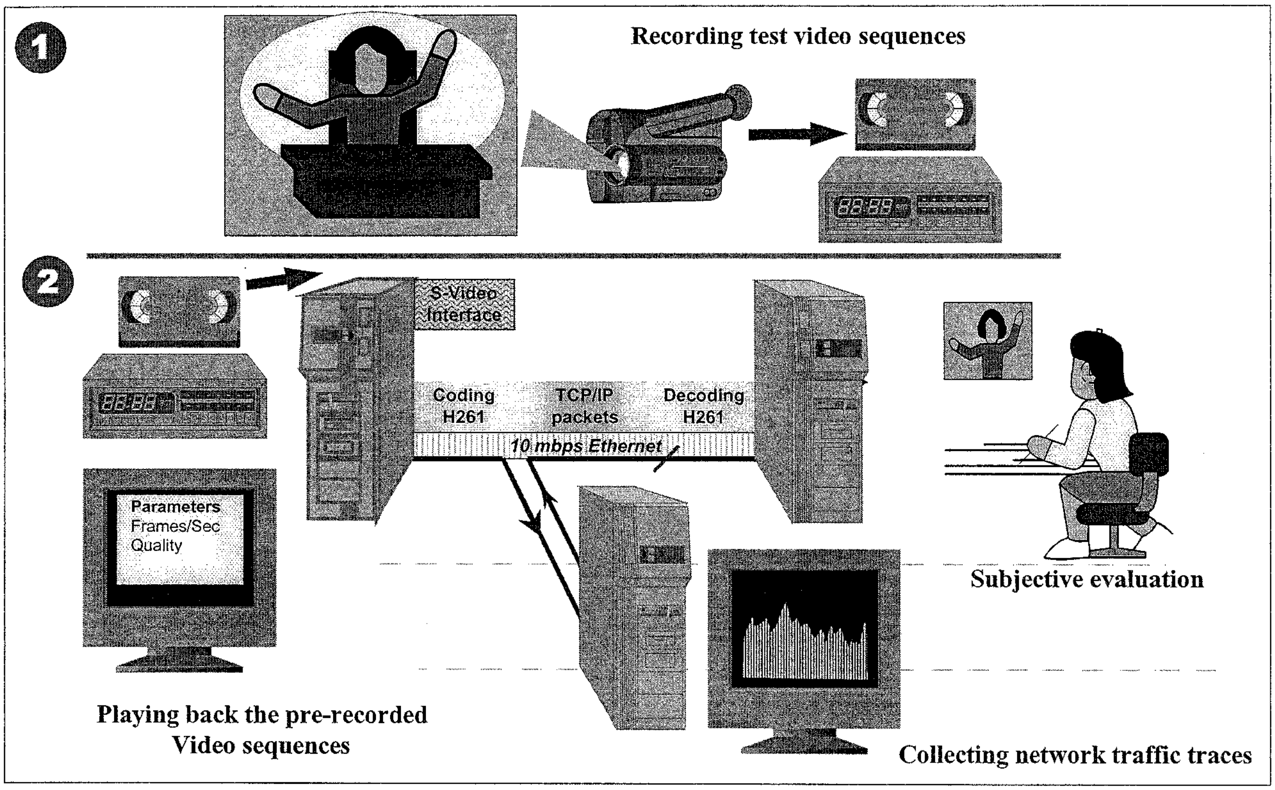

Fig. 5. Summary of the complete evaluation of a sign-language videotelephony service.

voice and picture, (e.g., lip sync) have comparable requirements. The attainable quality of the input scene is determined by a combination of ambient conditions—lighting and clothing and the quality of the capture equipment. The continuity and quality of the received picture depends on the ambient conditions and rate of playout combined with the codec algorithms, the end-to-end transmission mode, and the performance of the underlying network.

C. Equipment Configuration

There are many products offering videotelephony, using PCs as well as purpose built video telephones. The ITU's H and T series of recommendations are used widely for call control, coding, and framing. The choice was based on the assumption that IP will dominate.

Thus, those products that operate in packet mode, on local area network (LAN) connections and/or the Internet, are more relevant to the study reported here than those that operate over switched telephone networks, mainly narrow-band ISDN from 64 kb/s up to 2 Mb/s. They execute on mass-market computing platforms, using off-the-shelf video frame capture and display buffers and network interfaces. As we will explain, these have a profound effect on the behavior of the application, consequently on the data analysis.

The video codec H.261 [7] is the best established. Its video part encodes mainly changes in the stream of input frames, due to motion of objects in the scene. Implementations that were reviewed prior to this study appeared to adopt several different strategies for partitioning the scene; refreshing the static parts at different rates; or selecting blocks for update by different strategies. Anecdotal evidence suggests that these implementation choices are relatively unimportant by contrast with the influences exerted by the combined behavior of the execution platform, its software, and its hardware.

D. What Will be Measured and Why?

This study focuses on the *EB* parameter, α, as used in [9] and [10]. In this paper, α has been evaluated for real broad-band traffic data sets obtained from sign-language videotelephony for deaf users.

Two variable parameters have been considered: the frame rate of the video signal and the image quantization know as QUANT (macro-bloc factor in the H.261 terminology) and ranging from 1 (highest quantization) to 32 (lowest quantization).

Experiments were carried out for most combinations of frame rate (5, 6, 8, and 10 frame/s) and QUANT (9, 19, 25). For better clarity in the results, the values of QUANT will be associated hereafter as an image quality factor: *High* for 9, *Medium* for 19, and *Low* for 25.

E. Scenario

Fig. 5 represents a complete evaluation scenario of a sign-language videotelephony application over an Ethernet 10 Mb/s LAN. First, there is the recording of the test sign-language video clip on a VHS cassette. Then, by using a UNIX video conferencing package VIC [18], sequences from this clip were played back to a subject throughout the LAN with various values of frame rate and QUANT. In parallel, we were measuring the traffic that was circulating over the LAN.

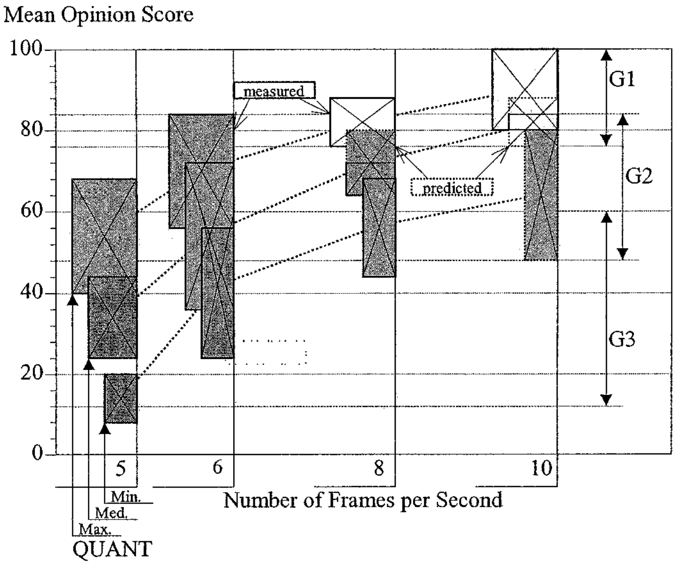

Fig. 6. Subjective assessment of the sign-language videotelephony application. Each shaded rectangle corresponds to a particular frame rate and QUANT value.

The total duration of this video clip is 160 s. It contains five sign-language sequences interleaved for a few seconds with inactive signing.

The picture format is common intermediate format (CIF), whose luminance sampling structure is 352 pixels per line and 288 lines per picture in an orthogonal arrangement.

V. Subjective Analysis

The purpose of a subjective analysis is to determine the values of quality parameters that correspond to thresholds of user's acceptability of a service under known conditions.

A. Collection and Quantification of Subjective Data

After each video sequence, subjects were asked to answer to four questions about the content of the video clip.

During the trials, assessors noticed that the subject's perception of the quality of a given service was biased by assumptions about the technology, or even by opinions about the individual's skill in communicating. In order to minimize the influence of these subjective parameters, the assessment focuses on the level of comprehension rather than on the degree of satisfaction with the service. In addition, levels of comprehension are scalable and easier to analyze.

Therefore, the assessment of all these questionnaires allowed determining an MOS in the range of 0–100.

B. Assessment Method and Results

Individual reactions from the subjects led to scattered subjective quality assessment. Involving twice as many subjects would significantly increase the confidence in the results. Nevertheless, even with ten subjects, a correlation between the quality of the video and the degree of comprehension has been identified and classified into three fuzzy groups:

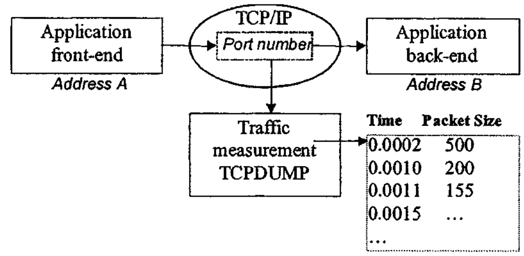

Fig. 7. Synopsis of the collection of datagrams on an Ethernet network.

- G1: Most of the subjects answered correctly at least 75% of the questions with some of them reaching high scores (the lightest area in Fig. 6).
- G2: The achievements were random, sometimes lower than 50% and no subject reached scores above 85%. Difficulties often arose from lip reading.
- G3: Both lip reading and signs were sometimes difficult to interpret. Scores ranged between 15% and 60%. (the darkest area in Fig. 6).

The result of the subjective assessment that is represented in Fig. 6 lies within the boxes. Three of the twelve combinations of frame rate and QUANT that we have considered corresponds to the highest MOS (G1) and three corresponds to the lowest MOS (G2). The six others are in the area of average MOS (G2).

VI. Objective Analysis

A. Collection of Traffic Traces

The most accurate method to determine the bandwidth usage of an application is to measure the traffic flow directly at the output of the network driver, but, in practice, this would require an application dependent measurement tool.

An alternative solution consists of monitoring the activity at the physical layer of the network in the absence of other sources in this particular TCP port.

This solution can be valid as long as the medium speed is higher than the maximum traffic peak generated by the application; otherwise significant time measurement errors will be introduced due to the packet dwell time in buffers.

For the case that we are considering, the time scale of the network bit rate being much smaller than the application time scale, the time measurement errors are assumed negligible.

Fig. 7 illustrates the method for collecting traffic traces using TCPDUMP [19]. The stream of packets is stored as a data set in the form of series of time-stamp and size pairs.

The chronograms (see Fig. 8) represent the instantaneous bandwidth required over an integration period of 1 s.

B. Analysis

The results displayed in Fig. 9 correspond to the EB surface $\alpha(s,t)$ for the 12 collected traffic traces.

As stipulated in the *property (iv)* of Kelly's equation, we can observe that, for a fixed value of t, α always increases with s. Nevertheless, minima occur along t for a fixed value of s,

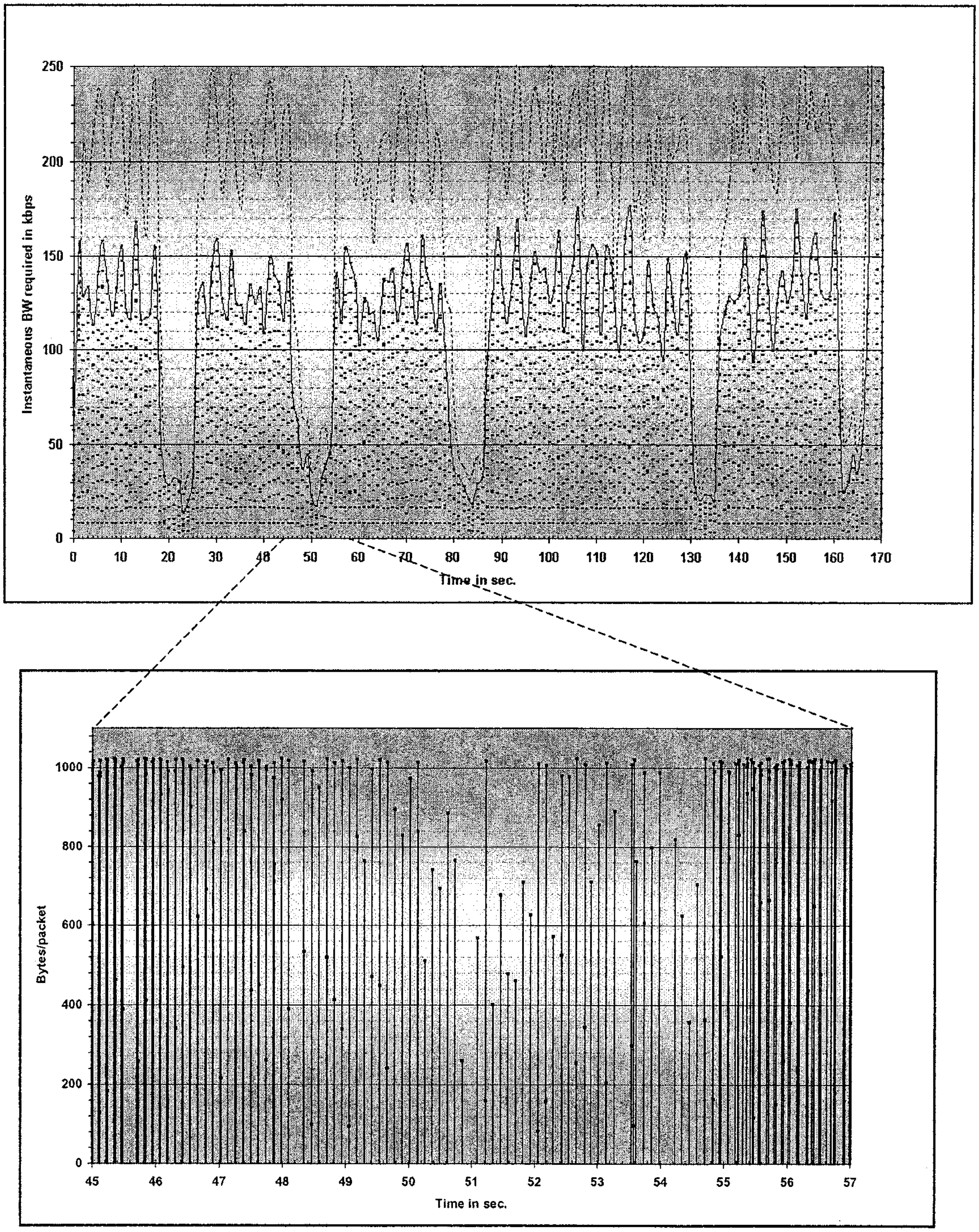

Fig. 8. Views of the cumulated macroscopic (above) and the microscopic (below) for a typical traffic trace. On the macroscopic view, each dot corresponds to a packet, the solid line is the envelope of the shaped traffic for 8 frames/s and the dashed line is the variation of traffic when the image resolution is increased. On the microscopic view, one may notice the almost constant packet size and the regular burst of packets every frame period (125 ms).

which is one of the properties of the on-off periodic sources analyzed in Kelly's paper [9]. One of these minima lies in any case between the frame rate period (FP) and half of this period (FP/2), Moreover, this pattern seems to repeat itself for larger values of t. This is particularly noticeable for the case 8 frames/s and medium QUANT.

For delay-sensitive applications like video, it is essential to keep t to a minimum value and s to a maximum value in order to minimize jitter effects. Thus for given constrains of jitter (expressed as a ratio of FP) and network capacity, these graphs can provide the optimum values of s and t.

VII. Example of Exploitation of These Results

From Fig. 10, let us consider one of the cases where the evaluation is close to an acceptable subjective quality. We choose the case with 8 frames/s and medium QUANT. If we suppose that the jitter effect will not affect the perceived quality when

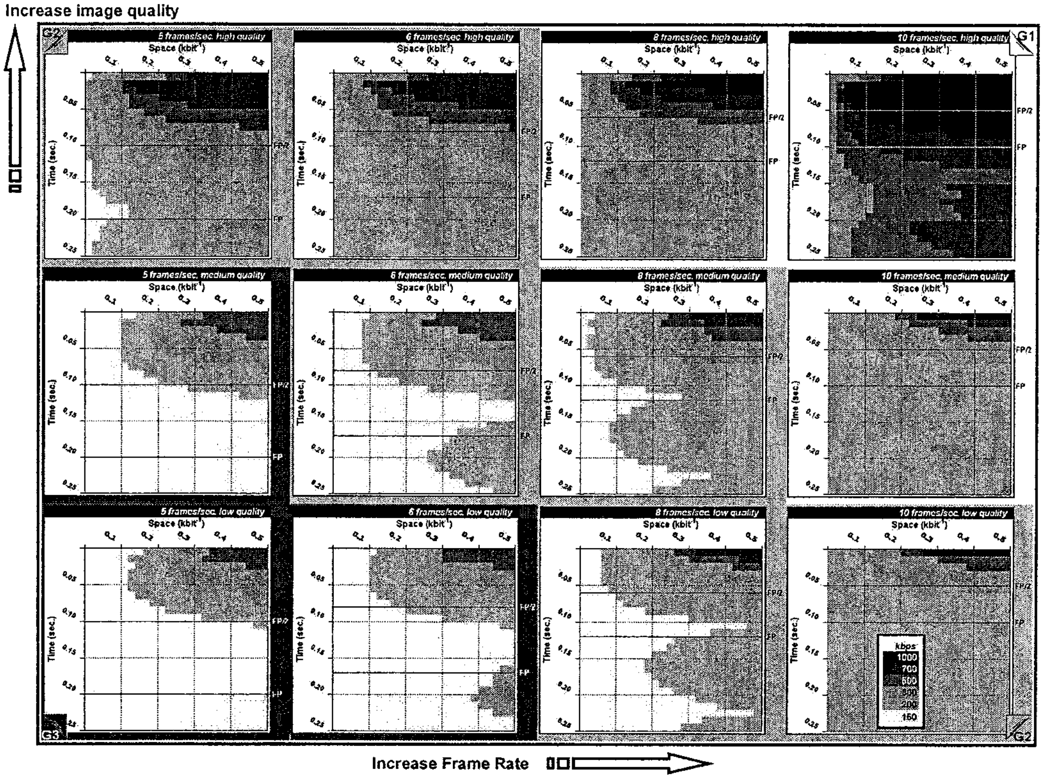

Fig. 9. EB surfaces of an H.261 sign-language video sequence for different frame rate (horizontal axis) and different values of QUANT (vertical axis). The results of the subjective evaluation are also represented in three different level of shaded areas. G1 in light gray at the top right of the graph, G2 in medium gray from the top left to the bottom right, and G3 in darker gray in the bottom left.

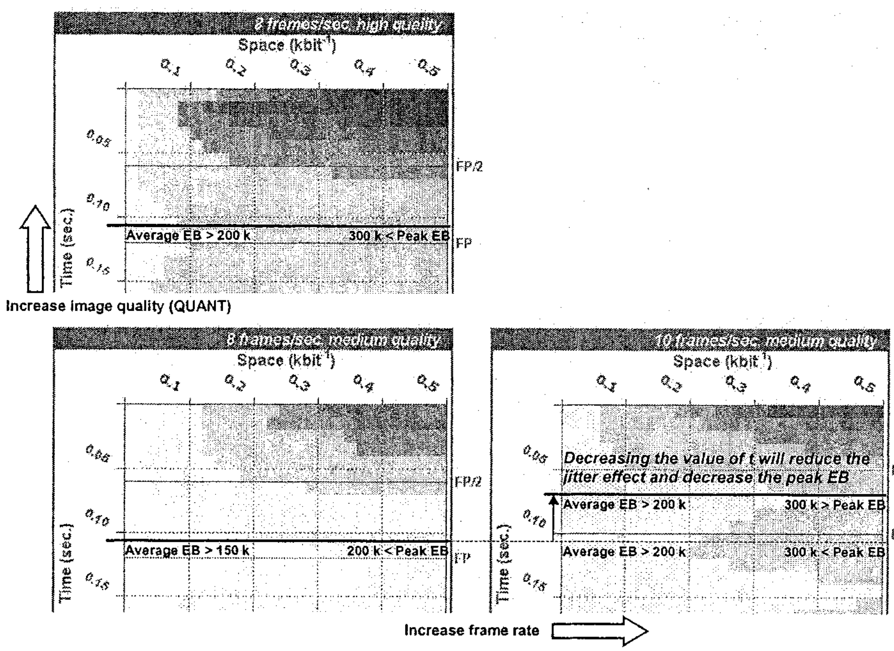

Fig. 10. Illustration of a practical exploitation of the EB analysis when statistical variation in the source or in the network capacity occurs. Note the pertinent change of t when the frame rate increases from 8 to 10 frames/s.

the traffic shaping factor t is below the frame rate, then a value $t = 0.12$ s will satisfy the quality requirements for an EB distribution characterized with an average of 150 kb/s (small s) and peaks lower than 300 kb/s (high s).

Thus, mechanisms of decision can determine whether the frame rate (a) or QUANT (b) can be increased if the network offers more capacity and the optimum t recalculated accordingly (c).

VIII. Conclusions

In this paper, we explained the methodological approach which was applied during our experiments to correlate objective measurements with subjective perception.

By means of our case study, we have specified the minimum quality, expressed in terms of EB, required by the deaf people when using a H.261 sign-language videotelephony application over the Internet.

The results of the subjective analysis show that the threshold of user acceptability (75% of the MOS) is achieved at 8 frames/s with medium image resolution. At this point, increasing the frame rate or the image resolution does not improve very much the user perception while increasing rapidly the required EB.

The results of the objective analysis show that there are optimum value for QoS mechanism parameters that compromise the user acceptability of the service with the requirements in terms of EB.

This paper contributes toward the understanding of the important issues before envisaging modeling the traffic of an application.

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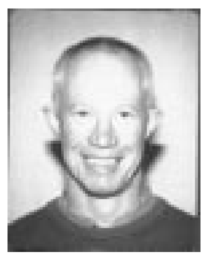

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